CLAIMS

1. A method of processing sound data, in which:

a) signals representative of at least one sound propagating in a three-dimensional space and arising from a source situated at a first distance (ρ) from a reference point (O) are coded so as to obtain a representation of the sound by components (B_{mn}^{σ}) expressed in a base of spherical harmonics, of origin corresponding to said reference point (O),

b) and a compensation of a near field effect is applied to said components (B_{mn}^{σ}) by a filtering which is dependent on a second distance (R) defining substantially, for a playback of the sound by a playback device, a distance between a playback point (HP_i) and a point (P) of auditory perception.

2. The method as claimed in claim 1, in which, said source being far removed from the reference point (O),

- components of successive orders m are obtained for the representation of the sound in said base of spherical harmonics, and
- a filter ($1/F_m$) is applied, the coefficients of which, each applied to a component of order m, are expressed analytically in the form of the inverse of a polynomial of power m, whose variable is inversely proportional to the sound frequency and to said second distance (R), so as to compensate for a near field effect at the level of the playback device.

3. The method as claimed in claim 1, in which, said source being a virtual source envisaged at said first distance (ρ),

- components of successive orders m are obtained for the representation of the sound in said base of spherical harmonics, and
- a global filter (H_m) is applied, the coefficients of which, each applied to a component of order m, are

expressed analytically in the form of a fraction, in which:

- the numerator is a polynomial of power m, whose variable is inversely proportional to the sound frequency and to said first distance (ρ), so as to simulate a near field effect of the virtual source, and
- the denominator is a polynomial of power m, whose variable is inversely proportional to the sound frequency and to said second distance (R), so as to compensate for the effect of the near field of the virtual source in the low sound frequencies.

4. The method as claimed in one of the preceding claims, in which the data coded and filtered in steps a) and b) are transmitted to the playback device with a parameter representative of said second distance (R/c).

5. The method as claimed in one of claims 1 to 3, in which, the playback device comprising means for reading a memory medium, the data coded and filtered in steps a) and b) are stored with a parameter representative of said second distance (R/c) on a memory medium intended to be read by the playback device.

6. The method as claimed in one of claims 4 and 5, in which, prior to a sound playback by a playback device comprising a plurality of loudspeakers disposed at a third distance (R_2) from said point of auditory perception (P), an adaptation filter ($H_m^{(R1/c,\ R2/c)}$) whose coefficients are dependent on said second (R_1) and third distances (R_2) is applied to the coded and filtered data.

7. The method as claimed in claim 6, in which the coefficients of said adaptation filter ($H_m^{(R1/c,R2/c)}$), each applied to a component of order m, are expressed

analytically in the form of a fraction, in which:

- the numerator is a polynomial of power m, whose variable is inversely proportional to the sound frequency and to said second distance (R),
- and the denominator is a polynomial of power m, whose variable is inversely proportional to the sound frequency and to said third distance (R_2).

8. The method as claimed in one of claims 2, 3 and 7, in which, for the implementation of step b), there is provided:

- in respect of the components of even order m, audiodigital filters in the form of a cascade of cells of order two; and
- in respect of the components of odd order m, audiodigital filters in the form of a cascade of cells of order two and an additional cell of order one.

9. The method as claimed in claim 8, in which the coefficients of an audiodigital filter, for a component of order m, are defined from the numerical values of the roots of said polynomials of power m.

10. The method as claimed in one of claims 2, 3, 7, 8 and 9, in which said polynomials are Bessel polynomials.

11. The method as claimed in one of claims 1, 2 and 4 to 10, in which there is provided a microphone comprising an array of acoustic transducers arranged substantially on the surface of a sphere whose center corresponds substantially to said reference point (O), so as to obtain said signals representative of at least one sound propagating in the three-dimensional space.

12. The method as claimed in claim 11, in which a global filter is applied in step b) so as, on the one hand, to compensate for a near field effect as a

function of said second distance (R) and, on the other hand, to equalize the signals arising from the transducers so as to compensate for a weighting of directivity of said transducers.

13. The method as claimed in one of claims 11 and 12, in which there is provided a number of transducers that depends on a total number of components chosen to represent the sound in said base of spherical harmonics.

14. The method as claimed in one of the preceding claims, in which in step a) a total number of components is chosen from the base of spherical harmonics so as to obtain, on playback, a region of the space around the point of perception (P) in which the playback of the sound is faithful and whose dimensions are increasing with the total number of components.

15. The method as claimed in claim 14, in which there is provided a playback device comprising a number of loudspeakers at least equal to said total number of components.

16. The method as claimed in one of claims 1 to 5 and 8 to 13, in which:

- there is provided a playback device comprising at least a first and a second loudspeaker disposed at a chosen distance from a listener,
- a cue of awareness of the position in space of sound sources situated at a predetermined reference distance (R) from the listener is obtained for this listener, and
- the compensation of step b) is applied with said reference distance substantially as second distance.

17. The method as claimed in one of claims 1 to 3 and 8 to 13, taken in combination with one of claims 4 and

5, in which:

- there is provided a playback device comprising at least a first and a second loudspeaker disposed at a chosen distance from a listener,
- a cue of awareness of the position in space of sound sources situated at a predetermined reference distance (R_2) from the listener is obtained for this listener, and
- prior to a sound playback by the playback device, an adaptation filter ($H_m^{(R/c,R2/c)}$), whose coefficients are dependent on the second distance (R) and substantially on the reference distance (R_2), is applied to the data coded and filtered in steps a) and b).

18. The method as claimed in one of claims 16 and 17, in which:

- the playback device comprises a headset with two headphones for the respective ears of the listener, and
- separately for each headphone, the coding and the filtering of steps a) and b) are applied with regard to respective signals intended to be fed to each headphone, with, as first distance (ρ), respectively a distance (r_R, r_L) separating each ear from a position (M) of a source to be played back.

19. The method as claimed in one of the preceding claims, in which a matrix system is fashioned, in steps a) and b), said system comprising at least:

- a matrix (B) comprising said components in the base of spherical harmonics, and
- a diagonal matrix ($\mathrm{Diag}(1/F_m)$) whose coefficients correspond to filtering coefficients of step b),

and said matrices are multiplied to obtain a result matrix of compensated components ($\tilde{B}$).

20. The method as claimed in claim 19, in which:

- the playback device comprises a plurality of loudspeakers disposed substantially at one and the same

distance (R) from the point of auditory perception (P), and

- to decode said data coded and filtered in steps a) and b) and to form signals suitable for feeding said loudspeakers:

  * a matrix system is formed comprising said result matrix ($\tilde{B}$) and a predetermined decoding matrix (D), specific to the playback device, and
  * a matrix (S) is obtained comprising coefficients representative of the loudspeakers feed signals by multiplication of the matrix of the compensated components ($\tilde{B}$) by said decoding matrix (D).

21. A sound acquisition device, comprising a microphone furnished with an array of acoustic transducers disposed substantially on the surface of a sphere, characterized in that it furthermore comprises a processing unit arranged so as to:

- receive signals each emanating from a transducer,
- apply a coding to said signals so as to obtain a representation of the sound by components (B_{mn}^{σ}) expressed in a base of spherical harmonics, of origin corresponding to the center of said sphere (O),
- and apply a filtering to said components (B_{mn}^{σ}), which filtering is dependent, on the one hand, on a distance corresponding to the radius of the sphere (r) and, on the other hand, on a reference distance (R).

22. The device as claimed in claim 21, characterized in that said filtering consists, on the one hand, in equalizing, as a function of the radius of the sphere, the signals arising from the transducers so as to compensate for a weighting of directivity of said transducers and, on the other hand, in compensating for a near field effect as a function of a chosen reference distance (R), defining substantially, for a playback of the sound, a distance between a playback point (HP_i) and a point (P) of auditory perception.